IN THE CLAIMS

1. (currently amended) An end-to-end estimation of the bandwidth available in a client-server connection established over a packet switching network, comprising:

a routine to compute samples of available bandwidth by taking into account the flow of data packets received by the client and ~~their arrival times~~ time intervals during which the data packets are received[[,]] if the routine is implemented at the client receiver side, or by taking into account acknowledgments or report packets received by the server sender side and ~~their arrival times~~ time intervals during which acknowledgment or report packets are received[[,]] if the routine is implemented at the server sender side;

a routine to compute ~~bandwidth~~ samples of available bandwidth as the ratio of the amount of received data packets over the time interval during which the data packets are received if the routine is implemented at the client receiver side, or as the ratio of the amount of the data packets ~~acked~~ acknowledged over the time interval during which the data packets are ~~acked~~ acknowledged if the routine is implemented at the server sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of the available bandwidth.

2. (currently amended) The end-to-end bandwidth estimation according to claim 1, wherein a sample of available bandwidth b_j at time t_j is computed as:

$$b_j = \frac{d_j}{t_j - t_{j-1}}$$

where d_j is the amount of data that have been received at the client receiver side or acknowledged at the server sender side in the interval $t_j - t_{j-1}$, t_{j-1} is the time when the previous ACK or the ACK one or more congestion windows of packets before ~~was~~ were received by the server sender side or the time when the previous packet or the packet one or more congestion windows of packets before ~~was~~ were received by the client receiver side , and t_j is the time when the current ACK is received by the server sender side or when the current packet is received by the client receiver side.

3. (original) The end-to-end bandwidth estimation according to claim 1, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.

4. (currently amended) The end-to-end bandwidth estimation according to claim 2, wherein the available bandwidth samples are computed according to claim 2 and are averaged using the discrete-time low-pass filter with time-varying coefficients:

$$\hat{b}_j = \frac{2\tau_f - \Delta_j}{2\tau_f + \Delta_j}\hat{b}_{j-1} + \Delta_j \frac{b_j + b_{j-1}}{2\tau_f + \Delta_j}$$

where $\hat{b}_j$ is the filtered measurement of the available bandwidth at time $t = t_j$, $\hat{b}_{j-1}$ is the filtered measurement of the available bandwidth at time t_{j-1}, $\Delta_j = t_j - t_{j-1}$, $1/\tau_f$ is the cut-off frequency of the filter, b_j is the sample of the available bandwidth at time t_j, and b_{j-1} is the sample of the available bandwidth at time t_{j-1}[[.]]if a time t_j/m ($m \geq 2$) has elapsed since the last received ACK or packet without receiving any new ACK or packet, then the filter assumes the reception of a virtual sample $b_j=0$.

5. (original) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 1.

6. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 3.

7. (currently amended) Method for adaptively setting congestion window (cwnd) and slow start threshold (ssthresh) in the TCP/IP protocol comprising an end-to-end bandwidth estimation according to claim 1 to set the windows as follows:

after a timeourt: set ssthresh=min(2, BWE*RTTmin)[[,]]

set cwnd=2;

after 3 dupack: set ssthresh=min(2, BWE*RTTmin)[[,]]

set cwnd=ssthresh; and

wherein RTT min is the minimum round trip time and BWE is the available bandwidth computed according to claim 1 at the time of timeout or when 3 dupacks or n are received.

8. (currently amended) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end ~~bandwidth~~ estimation ~~comprising:~~

~~a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and~~

~~a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,~~

~~wherein the low-pass filter is a low-pass filter according to claim 3~~ <u>of the bandwidth available according to claim 1</u>.

9. (currently amended) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol, comprising<u>:</u> ~~an end-to-end bandwidth estimation according to claim 1~~

<u>a routine to compute an end-to-end estimation of the available bandwidth according to claim 7;</u>

<u>a routine that selects the quality of coding, or the nembers of layers to be transmitted in a layered coding of an audio/video so that the sending rate is the closer to the end-to-end bandwidth available estimated according to claim 7; and</u>

<u>a routine to set TCP congestion window and slow start threshold according to claim 7 in order to send the coded audio/video source</u>.

10. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low pass filter is a low pass filter according to claim 3.

11. (currently amended) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source ~~according to claim 9~~, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the <u>available</u> bandwidth estimation <u>at time of congestion according to claim 1</u> ~~comprising:~~

~~a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into~~

~~account acknowledgment packets received by the sender, if the routine is implemented at the sender side;~~

~~a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth~~ and[[;]] increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth <u>until a congestion episode is experienced again</u>.

12. (currently amended) Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the <u>available</u> bandwidth estimate times the minimum round trip time, wherein the <u>available</u> bandwidth is computed according to claim 1.

13. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low pass filter is a low pass filter according to claim 4.

14. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

15. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

16. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source according to claim 10, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side;

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth; and

increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth.

17. (currently amended) Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the available bandwidth estimate times the minimum round trip time, wherein the ~~samples are~~ available bandwidth is computed according to claim 1 ~~2, and the bandwidth estimate is computed by:~~

~~a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into~~

~~account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and~~

~~a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth~~.

18. (canceled)

19. (previously presented) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 3.

20. (previously presented) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 4.

21. (new) The end-to-end bandwidth estimation according to claim 2, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.